What is claimed is:

1. A method, comprising:

detecting sound with a sensor to generate a corresponding sensor signal;

generating data with the sensor signal in accordance with a maximum likelihood estimator; and

filtering the data with an order-statistics filter to provide an estimate of reverberation time.

2. The method of claim 1, which includes iteratively determining a decay time parameter and a power parameter during execution of said generating.

3. The method of claim 1, which includes providing the reverberation time to one or more of a hearing assistance data processing routine, a voice recognition data processing routine, a hands-free telephony data processing routine, a teleconference data processing routine, and a sound level evaluation data processing routine.

4. The method of claim 1, wherein said generating includes calculating a number of reverberation time parameter estimations with the maximum likelihood estimator, the estimations each being calculated as a function of a sequence of sound observations over a different time window.

5. A method, comprising:

detecting sound with a sensor to generate a corresponding sound signal;

iteratively determining two or more values with a maximum likelihood function to evaluate one or more reverberation characteristics of an acoustic environment, one of the values corresponding to a time-constant parameter and another of the values corresponding to a diffusive power parameter; and

providing an estimate corresponding to reverberation time of the acoustic environment.

6. The method of claim 5, wherein said iteratively determining is performed for each of a number of different frequency ranges of the sound and includes calculating a reverberation time estimate for each of the different frequency ranges.

7. The method of claim 5, which includes providing the estimate of reverberation time to one or more of a hearing assistance data processing routine, a voice recognition data processing routine, a hands-free telephony data processing routine, a teleconference data processing routine, and a sound level evaluation data processing routine.

8. The method of claim 5, wherein said iteratively determining is performed with each of a number of different data sequences to provide a number of reverberation time estimations, the data sequences each being representative of a series of sound observations over a different time window.

9. The method of claim 8, wherein said providing includes filtering the reverberation time estimations with an order-statistics filter to select the estimate corresponding to reverberation of the acoustic environment.

10. A method, comprising:

preparing a number of observed sound data sequences each representative of sound over one of a number of different time periods;

determining each of a number of estimations as function of a different one of the sequences in accordance with a maximum likelihood estimator; and

selecting one of the estimations to provide a reverberation time.

11. The method of claim 10, wherein said selecting includes filtering the estimations.

12. The method of claim 11, wherein said filtering is performed with an order-statistics filter.

13. The method of claim 10, wherein said determining includes iteratively calculating each of at least two values, a first one of the values corresponding to a decay time and a second one of the values corresponding to diffusive power.

14. The method of claim 13, wherein said iteratively calculating is performed for each of a number of different frequency ranges of the sound and includes calculating a reverberation time estimate for each of the different frequency ranges.

15. The method of claim 10, which includes providing the one of the estimations to one or more of a hearing assistance data processing routine, a voice recognition data processing routine, a hands-free telephony data processing routine, a teleconference data processing routine, and a sound level evaluation data processing routine.

16. The method of claim 10, wherein the sequences each correspond to a different series of observed sound samples and the different time periods each correspond to a different time window.

17. The method of claim 16, which includes adaptively changing the duration of the different time windows.

18. A method, comprising:

preparing a number of observed sound data sequences each corresponding to sound over a different one of a number of time windows;

determining a number of reverberation time parameter estimations, the estimations each being determined as a function of a different one of the sequences in accordance with a parameter estimator; and

filtering the estimations with an order-statistics filter.

19. The method of claim 18, wherein the parameter estimator is based on maximum likelihood estimation.

20. The method of claim 18, wherein the parameter estimator is of a robust type.

21. The method of claim 18, wherein said determining includes iteratively calculating each of at least two of the parameters for each of the sequences, a first one of the parameters corresponding to a decay time and a second one of the parameters corresponding to power.

22. The method of claim 21, wherein said iteratively calculating is performed for each of a number of different frequency ranges of the sound and includes calculating a reverberation time parameter estimate for each of the different frequency ranges.

23. The method of claim 18, which includes selecting one the estimations based on said filtering.

24. The method of claim 23, which includes providing the one of the estimations to one or more of a hearing assistance data processing routine, a voice recognition data processing routine, a hands-free telephony data processing routine, a teleconference data processing routine, and a sound level evaluation data processing routine.

25. The method of claim 18, which includes adaptively changing the duration of the time windows.

26. A system, comprising:

a sensor for detecting sound; and

a processing subsystem operable to receive sound-representative signals from the sensor to determine a reverberation time estimate of an unknown acoustic environment by processing the signals with a parameter estimator and order-statistics filter.

27. The system of claim 26, wherein the parameter estimator is of robust type.

28. The system of claim 26, wherein the parameter estimator is of a maximum likelihood type.

29. The system of claim 26, further comprising means for processing the reverberation time estimate in one or more of a hearing assistance data processing routine, a voice recognition data processing routine, a hands-free telephony data processing routine, a teleconference data processing routine, and a sound level evaluation data processing routine.

30. The system of claim 26, wherein the subsystem includes:

means for processing a number of sequences of observed sound data to provide a corresponding number of reverberation time parameter estimations with the parameter estimator, the sequences each corresponding to one of a number of different time windows; and

means for filtering the estimations with the order-statistics filter to provide the reverberation time estimate.

31. The system of claim 30, further comprising at least one of:

means for performing the processing means for each of a number of different frequency ranges of the sound; and

means for adaptively changing the duration of the time windows.

32. An apparatus, comprising: a device carrying logic executable by a processor to process data corresponding to a number of sequences of sound samples, the sequences each being representative of sound over a different time period, the logic being further operable to determine a number of reverberation time parameter estimations each as function of a different one of the

sequences in accordance with a maximum likelihood estimator and filter the estimations to provide a selected reverberation time estimate.

33. The apparatus of claim 32, wherein the logic includes a number of software instructions and the device includes a computer-readable memory storing the software instructions.

34. The apparatus of claim 32, wherein the device includes one or more parts of a computer network and the logic is encoded in one or more signals by the device.

35. The apparatus of claim 32, wherein the logic is operable to filter the estimations with an order-statistics type of filter.